

(19) World Intellectual Property Organization  
International Bureau



(43) International Publication Date  
12 April 2001 (12.04.2001)

PCT

(10) International Publication Number  
**WO 01/26418 A1**

(51) International Patent Classification<sup>7</sup>: H04R 25/00

(21) International Application Number: PCT/DK99/00531

(22) International Filing Date: 7 October 1999 (07.10.1999)

(25) Filing Language: English

(26) Publication Language: English

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(81) Designated States (*national*): AU, CA, JP, SG, US.

(84) Designated States (*regional*): European patent (AT, BE,  
CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC,  
NL, PT, SE).

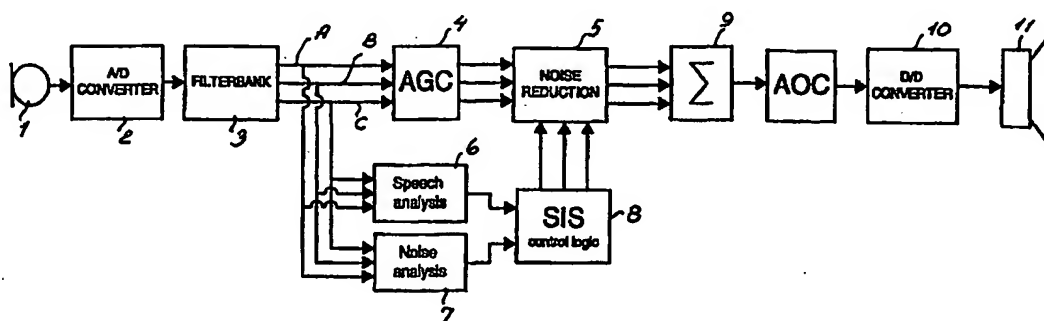
Published:

— With international search report.

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ance Notes on Codes and Abbreviations" appearing at the begin-  
ning of each regular issue of the PCT Gazette.

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(54) Title: METHOD AND SIGNAL PROCESSOR FOR INTENSIFICATION OF SPEECH SIGNAL COMPONENTS IN A HEARING AID



(57) Abstract: In a hearing aid signal processor with AGC in at least three processing channels (a, B, C) for different frequency bands and with noise squelching capability to affect the gain control in at least a lowest frequency band (A) and one intermediate frequency band (B) speech signals components in the intermediate frequency band of an input signal including background noise are intensified by estimation of the content of speech signal components in at least the highest frequency band (C) and modification of the gain adjustment caused by noise squelching in the intermediate frequency band (B) to reduce the noise squelching.

WO 01/26418 A1

**Method and signal processor for intensification of speech signal components in a hearing aid.**

The present invention relates to a method for  
5 intensification of speech signals components in a hearing aid input signal including background noise, comprising the steps of classifying said input signal into at least three frequency bands comprising at least one higher frequency band, a lowest frequency band and  
10 at least one frequency band intermediate said higher frequency band and said lowest frequency band, estimating the level of background noise in said lowest frequency band and at least one intermediate frequency band and adjusting the gain in said lowest and said one  
15 intermediate frequency bands in response to the estimated level of background noise to provide squelching of said background noise.

In WO 99/34642 a hearing aid having a signal processor with multiple processing channels is dis-  
20 closed, in which dynamic automatic gain control is effected by detection of the input sound level and/or the output sound level and adapting the output sound level in response to the detected sound level by controlling the gain in each processing channel towards an  
25 actually desired value of the output sound level. The gain control is effected at increases and decreases, respectively, of the input sound level by adjusting the gain towards the desired value with an attack time and a release time, respectively, which in response to the  
30 detected sound level are adjusted to a relatively short duration providing fast gain adjustment at high input and/or output sound levels and to a relatively long duration providing slow gain adjustment at low input and/or output sound levels.

35 In a practical implementation of this prior art

hearing aid the dynamic gain control is effected partly on the basis of the momentary sound input received by the hearing aid, partly on the basis of a statistical analysis of the sound level within a time window  
5 extending 20 to 30 seconds back in time. The actual gain adjustment is calculated by a complex algorithm to determine the actual gain control in each channel and the rate of control.

This dynamic gain control has appeared to offer  
10 significant advantages compared to known AGC methods for hearing aid gain control. At low sound levels, at which the transfer function provides a compressor characteristic and the reproduced sound is sensitive to pumping or vibrating sound effects at varying gain the  
15 sound will be controlled with long attack and release times, whereas at high sound levels at which the reproduced sound approaches the clipping or pain threshold the sound will be controlled with short attack and release times.

20 This prior art hearing aid has moreover been implemented with an effective noise suppression based on detection of the contents of speech and noise in each processing channel. In the absence of noise, the noise suppression or noise squelching is not effective, whereas  
25 at the occurrence of heavy noise in a frequency band the gain adjustment otherwise resulting from the dynamic gain control is modified towards a reduced gain. Thereby, the advantage is obtained that use of the hearing aid in a noisy environment in a relatively  
30 long time is made possible without causing unacceptable discomfort to the user.

In general, the use of temporary noise suppression or noise squelching in hearing aids or similar devices has been disclosed in several prior art publications.

35 US-A-4,630,302 discloses a method and apparatus

for aiding hearing with an automatic gain control unit having a first section for increasing the amplitude of input signal segments below a threshold level and a second section for reducing the amplitude of input  
5 signal segments above the threshold level. A noise suppressor unit having a long attack time and a short release time is responsive to the output from the second section of the automatic gain control unit and has a threshold level of operation below the threshold  
10 level of the automatic gain control unit to pass speech signals and squelch background noise signals between speech signal segments.

In US-A-4,852,175 a hearing aid signal processing system is disclosed, in which noise squelching in each  
15 of a plurality of frequency bands is effected by estimation of the absolute quantity of noise by monitoring the amplitude distribution of sound events in each band and comparing the absolute quantity of noise in a current frequency band, in which gain is to be adjusted,  
20 with the absolute quantity of noise in a next higher frequency band, whereby the gain in the current frequency band is reduced, if the noise quantity in this band exceeds the noise quantity in the next higher band by more than a predetermined threshold value.

25 In US-A-5,768,473 an adaptive speech filter is disclosed, in which frequency components of an information signal from an input signal also containing noise is effected by calculation of the total power in each frequency component, estimating the power of the infor-  
30 mation signal included therein and calculating a modified gain for each frequency band as a function of the total power, the information signal power estimate and a previous estimate of a noise power, the input frequency component being multiplied by said modified gain  
35 to produce an estimate of the power the frequency com-

ponent of the information signal and a new noise power estimate being estimated from the previous noise power estimate and the difference between the total power in the frequency component and the estimate of the power  
5 of the frequency component of the information signal, regardless of whether there is a pause in the information signal.

In the noise squelching implemented in the prior art hearing aid of WO 99/34642 the statistical noise  
10 estimation in each frequency band will result in a relatively slow gain reduction, which in case of input signals containing speech and noise components having comparable sound levels has been observed to reduce the perception of speech in certain situations, e.g. when  
15 the hearing aid is used during car driving.

On this background, it is the object of the invention to provide a signal processing method and a signal processor for a hearing aid, in which the content of speech in an input signal also containing noise is  
20 intensified to improve the perception of speech.

To meet this object the speech intensification method of the invention is characterized by the steps of estimating the content of speech signal components in said at least one said higher frequency band and  
25 modifying the gain adjustment caused by said background noise estimation in at least said one intermediate frequency band to reduce said squelching of background noise and thereby intensify the content of speech signal components occurring in said one intermediate  
30 frequency band.

The invention is based on the recognition of the fact that the observed reduction in speech intelligibility referred to above is caused by the effect of upward spread or masking of noise, by which noise  
35 typically occurring in the lowest frequency band of the

signal processing system is spread upwards to the next higher intermediate frequency band, which will normally contain frequency components of significant importance for speech perception. In result, noise squelching will  
5 be effected not only in the lowest frequency band having the major content of noise, but also in the next higher intermediate frequency band.

By the modification of the gain adjustment in this frequency band on the basis of speech components in at  
10 least the highest frequency band, in accordance with the invention, an intensification of the speech content in intermediate frequency band is effected, which has been observed to provide significant improvement of speech perception.

15 Whereas the estimation of noise and speech signal components can be effected by a variety of methods known per se, such as disclosed in WO 99/34642, e.g. FFT analysis or peak detection, it is preferred for the method according to the invention that following said  
20 classification said input signal is subjected to digital signal processing in each of said frequency bands and said estimations of the quantity of background noise and the content of speech signal components are effected by percentile estimation.

25 For the performance of the speech intensification method of the invention a signal processor for a hearing aid is provided, which comprises means for receiving an input signal containing speech signal components in the presence of background noise, means  
30 for classification of said input signal into at least three frequency bands comprising at least one higher frequency band, a lowest frequency band and at least one frequency band intermediate said higher frequency band and said lowest frequency band, variable gain  
35 adjustment means for controlling the gain in each of

said frequency bands and means for estimating the level of background noise in said lowest frequency band and at least one intermediate frequency band and adjusting the gain in said lowest and said one intermediate  
5 frequency bands in response to the estimated level of background noise to provide squelching of said background noise.

According to the invention, the signal processor is characterized in that means is provided for estimating the content of speech signal components in said at  
10 least one higher frequency band and for modifying the gain adjustment caused by said background noise estimation in at least said one intermediate frequency band to reduce said squelching of background noise and  
15 thereby intensify the content of speech signals components occurring in said one intermediate frequency band.

In the following, the invention will be explained in further detail with reference to the accompanying  
20 drawings, in which

fig. 1 is an exemplified schematic block diagram of a 3-channel hearing aid signal processor embodying the invention;

figs. 2 and 3 are graphic representations of sound  
25 level as a function of frequency for typical speech and noise components of a combined sound input signal received by the signal processor in fig. 1;

figs. 4 and 5 are graphic representations of filter damping/attenuation for an input sound signal composed of the speech and noise components as illustrated  
30 in fig. 2 and 3 by use of conventional noise squelching and by speech intensification according to the invention, respectively;

figs. 6 and 7 are graphic representations of the  
35 effect on the output signal level of the AGC and noise

sqelching illustrated in figs. 4 and 5, respectively, and

figs. 8 to 10 are graphic resprentations of typical amplitude distributions for speech, noise and a combination of speech and noise.

In fig. 1 a 3-channel hearing aid with digital signal processing is shown, in which sound input signals received by a microphone 1 are supplied to an A/D converter 2, the digital output signal of which is supplied to a filter bank 3, by which the digital signals are distributed in three frequency bands comprising a lowest frequency band, an intermediate frequency band and a highest frequency band as denoted by the three output lines A, B and C from filter bank 3.

For each frequency band a separate processing channel A, B and C, respectively, is provided. As shown in the figure these processing channels A, B and C are similar in structure and each includes a series arrangement of AGC controlled amplification means 4 and a noise reduction or noise sqelching block 5.

In each channel, the relevant output signal from the filter bank 3 is supplied in parallel to speech analyzing means 6 and noise analyzing means 7 supplying output signals to a speech intensification or SIS control logic block 8, from which control signals can be supplied to the noise sqelching block 5 in the respective processing channel A, B or C.

The digitally processed output signal from each of channels A, B and C is supplied via a summation device 9 and a D/A or D/D converter 10 to an output sound transducer 11 such as a loud speaker.

From the graphic representation in fig. 2 of the amplitude versus frequency relationship for e typical speech signal it can be seen, that a significant part

of the sound energy in the speech signal will be located in the intermediate frequency band B, typically ranging from 800 Hz to 2500 Hz, and that also a detectable portion of the sound energy will occur in the  
5 high frequency band C.

From the graphic representation in fig. 3 of a typical frequency spectrum for car noise, as perceived by a person inside the car, it is seen on one hand that the dominant part of the sound energy will be present  
10 in the lowest frequency band A.

The graphic representation in fig. 4 illustrates the effect on the normal gain control, e.g. by AGC, of a hearing aid provided with a conventional noise squelching system as explained above and receiving a  
15 sound input signal composed of the speech and noise components illustrated in figs. 2 and 3. The three columns indicate the increase of filter damping of a gain controlling filter in each of the three processing channels A, B and C caused by noise squelching compared  
20 to the damping caused by the normal gain control means of the hearing aid for a sound input signal containing the speech component only, i.e. without any noise component.

As illustrated, practical experience has shown  
25 that in case of a sound input signal containing speech as well as noise components. e.g. as illustrated in figs 2 and 3, the filter damping will be significantly increased not only in the lowest frequency band, where the dominant part of the noise energy is present, but  
30 also in the intermediate band, even if the noise energy in that band in the sound input signal in many situations, like the specific example of car noise, is significantly smaller than in the lowest frequency band. As explained above, this phenomenon is caused by  
35 an upwards spread or masking effect from the lowest

frequency band to the intermediate frequency band and results in a significant damping also of speech signal components in this band, whereby the perception of speech in the output sound signal from transducer 11 will decrease significantly for the majority of hearing impaired users.

By means of the method and signal processor of the invention this disadvantage can be substantially reduced. As shown in fig. 1, each of three processing channels A, B and C comprises in addition to the noise analyzing means 7, as used per se in known noise squelching systems, speech analyzing means 6 for detection and analyzing of the content of speech in the frequency band supplied to the respective processing channel.

In view of the normal spectral distribution of noise, as illustrated e.g. in fig. 3, it may strictly spoken only be of advantage to detect and analyze the content of speech in the high frequency band C, but for obvious structural reasons each of processing channels A, B and C, which are normally fully implemented as integrated circuits, should be similar in structure.

The output signals from the speech and noise analyzing means 6 and 7 in each of processing channels A, B and C are supplied to SIS control logic block 8, which in response will supply control signals to the noise squelching block 5 in the respective processing channel A, B or C.

The operation can be explained as follows.

For a sound input signal comprising speech without noise, i. e. typically speech in quiet surroundings, neither the noise squelching nor the speech intensification capability of the signal processor will be in function and the normal AGC controlled amplification performance of the hearing aid will remain unaffected.

For a sound input signal consisting of noise only the noise components will be detected and analyzed by noise analyzing means 7, the output signal of which is supplied via SIS control logic 8 directly to the noise squelching block 5 in the processing channel or channels affected by the noise to effect conventional noise squelching as known in the art.

In case of a sound input signal comprising speech in the presence of noise as outlined above the detection of speech in the highest frequency band C will cause a modification of the noise squelching in the intermediate frequency band B, by which as shown in fig. 5 the increase of filter damping is lowered compared to conventional noise squelching otherwise resulting from the detection of noise.

Whereas fig. 6 illustrates the effect of conventional noise squelching as illustrated in fig. 4 on the sound output signal from transducer 11, fig. 7 shows a significant speech intensification in the intermediate frequency band B.

For a digital hearing signal processor as shown in fig. 1 the speech and noise analyzing means 6 and 7 are preferably combined and implemented in an integrated structure employing two percentile estimators 12 and 13. Such percentile estimators are known in principle from US-A-4,204,260 and their use for automatic gain control in hearing aids has been disclosed in WO 95/15668 as well as in WO 99/34642 quoted above, the disclosure of which is incorporated herein by reference.

For the purpose of the noise squelching and speech intensification capability of the method and hearing aid signal processor of the present invention the percentiles of percentile estimators 12 and 13 can be adjusted to figures between 5 and 40 % and between 60 and

95 %, e.g. to 10 % and 90 %, respectively.

From percentile detectors 12 and 13 output signals are supplied to SIS control logic block 8 indicating the amplitude levels forming upper limits for 10 % and 5 90 %, respectively, of the input signal analyzed by percentile estimators 12 and 13 within a time window of a duration of e.g. 25 seconds.

As illustrated in the histogram in fig. 8 the amplitude distribution of a typical pulse-type speech  
10 signal in a quiet environment covers a wide range of amplitude levels corresponding to a relatively large separation of the 10 % and 90 % percentiles, whereas the amplitude distribution of a typical continuous  
15 9 be confined in a rather narrow range of amplitude levels with much smaller separation of the 10 % and 90 % percentiles.

For an input signal containing speech in the presence of noise the amplitude distribution formed by  
20 overlapping of the histograms in figures 8 and 9 will as shown in the histogram in figure 10 form an intermediate between the two extremes of pure speech and pure noise.

This relationship can be used in a simple way by  
25 SIS control logic block 8 to effect the control of noise squelching block 5 and provide the speech intensification described above.

Whereas the invention has been explained in the foregoing with reference to a 3-channel hearing aid in  
30 which estimation of the content of speech signal components is effected in the highest frequency band, this is not limiting for the invention. In the case, for instance, of multi-channel hearing aids having more than three channels processing signals in a corresponding  
35 number of frequency bands the estimation of speech

12

signal components could be effected with the same advantage in any higher frequency band or combination of bands, for which speech signal components dominate over the noise level.

5

## P A T E N T   C L A I M S

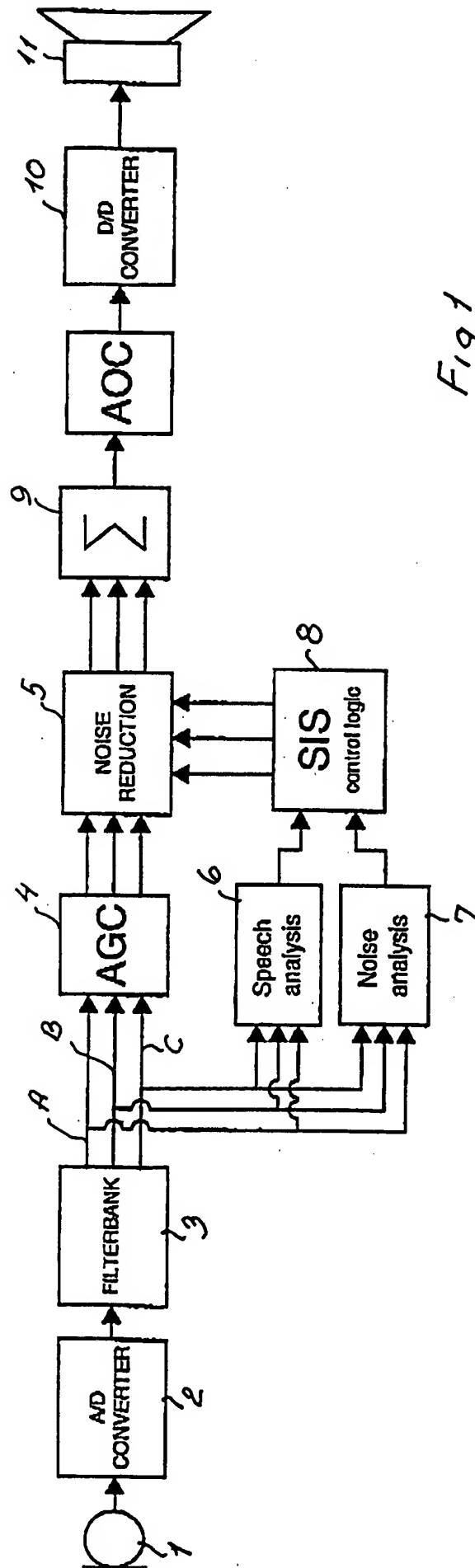
1. A method for intensification of speech signals components in a hearing aid input signal including background noise, comprising the steps of classifying  
5 said input signal into at least three frequency bands comprising at least one higher frequency band, a lowest frequency band and at least one frequency band intermediate said higher frequency band and said lowest frequency band, estimating the level of background  
10 noise in said lowest frequency band and at least one intermediate frequency band and adjusting the gain in said lowest and said one intermediate frequency bands in response to the estimated level of background noise to provide squelching of said background noise, c h a -  
15 r a c t e r i z e d by the steps of estimating the content of speech signal components in said at least one higher frequency band and modifying the gain adjustment caused by said background noise estimation in at least said one intermediate frequency band to  
20 reduce said squelching of background noise and thereby intensify the content of speech signals components occurring in said one intermediate frequency band.

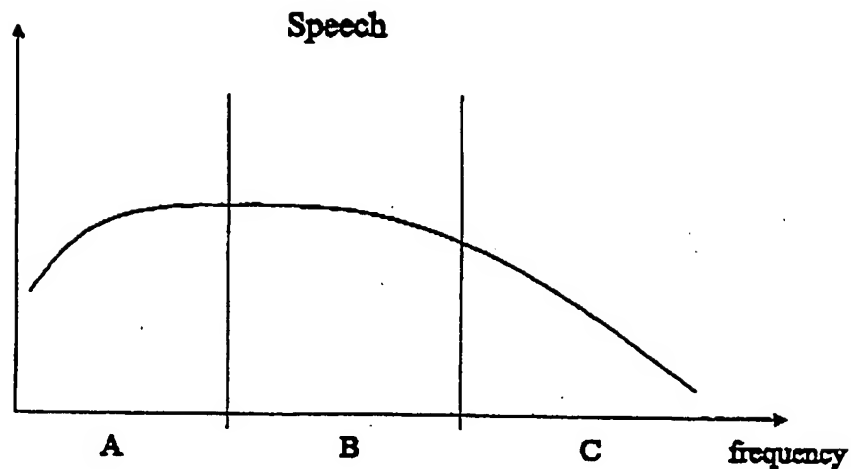
2. A method as claimed in claim 1, c h a r a c -  
t e r i z e d in that following said classification  
25 said input signal is subjected to digital signal processing in each of said frequency bands and said estimations of the level of background noise and the content of speech signal components are effected by percentile estimation.

30 3. A signal processor for a hearing aid, comprising means (1) for receiving an input signal containing speech signal components in the presence of background noise, means (3) for classification of said input signal into at least three frequency bands comprising  
35 at least one higher frequency band (C); a lowest

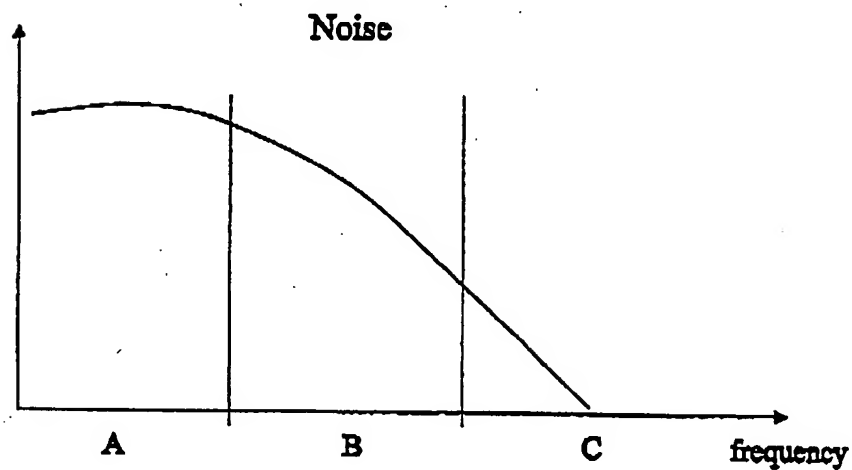
frequency band (A) and at least one frequency band (B) intermediate said higher frequency band and said lowest frequency band, variable gain adjustment means (4) for controlling the gain in each of said frequency bands  
5 and means (5, 7) for estimating the level of background noise in said lowest frequency band and at least one intermediate frequency band and adjusting the gain in said lowest and said one intermediate frequency bands in response to the estimated level of background  
10 noise to provide squelching of said background noise, characterized in that means (6, 8) is provided for estimating the content of speech signal components in said at least one higher frequency band and for modifying the gain adjustment caused by said  
15 background noise estimation in at least said one intermediate frequency band to reduce said squelching of background noise and thereby intensify the content of speech signal components occurring in said one intermediate frequency band.

20 4. A hearing aid signal processor as claimed in claim 3, characterized in that said estimation means comprises percentile estimator means (12, 13) for estimation of the level of background noise and the content of speech signal components.

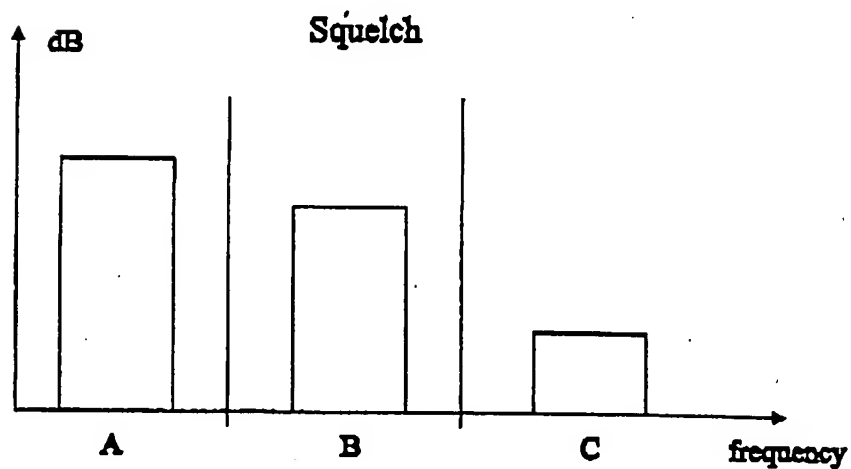




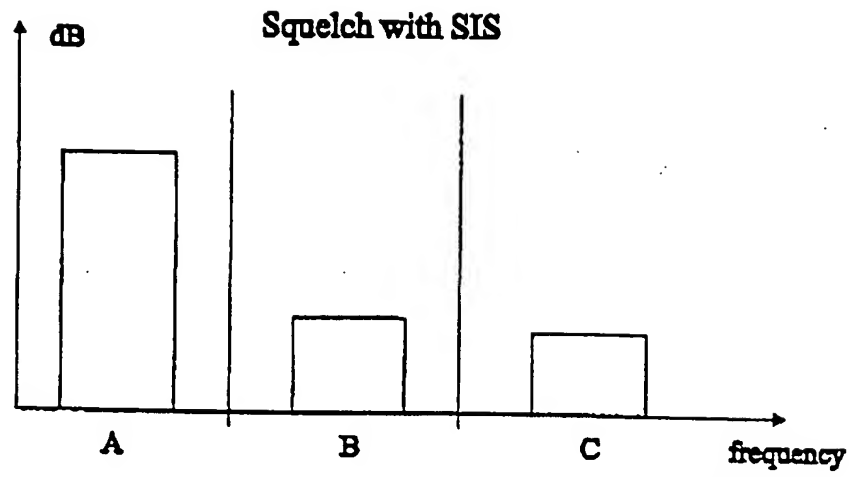
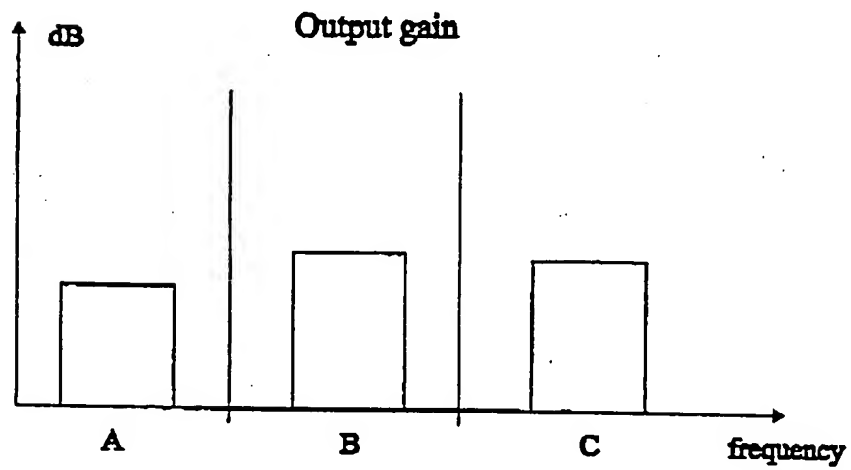
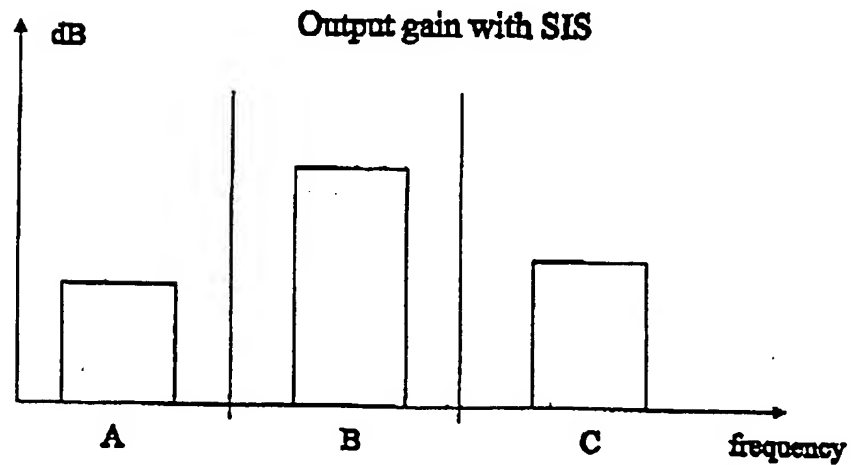
*Fig 2*



*Fig 3*



*Fig 4*

*Fig 5**Fig 6**Fig 7*

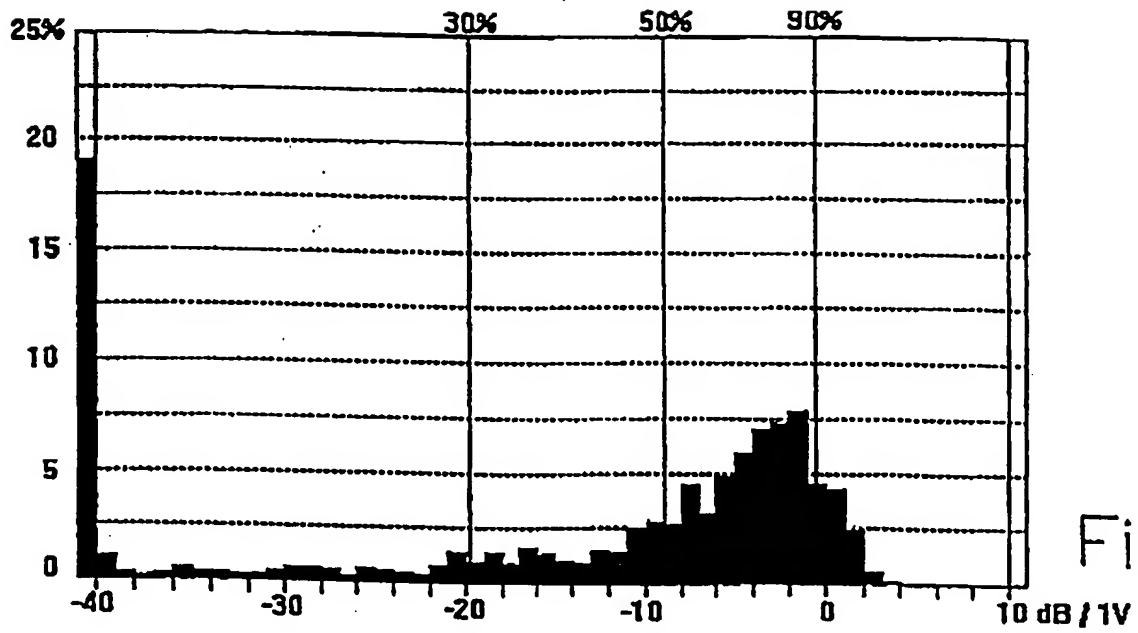


Fig.8

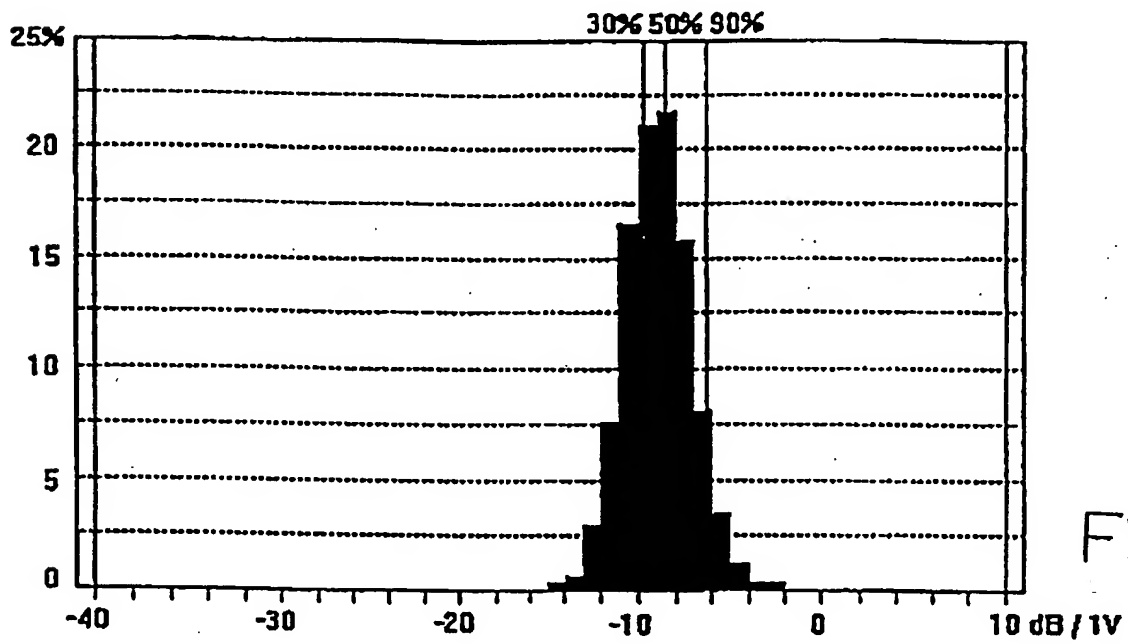


Fig.9

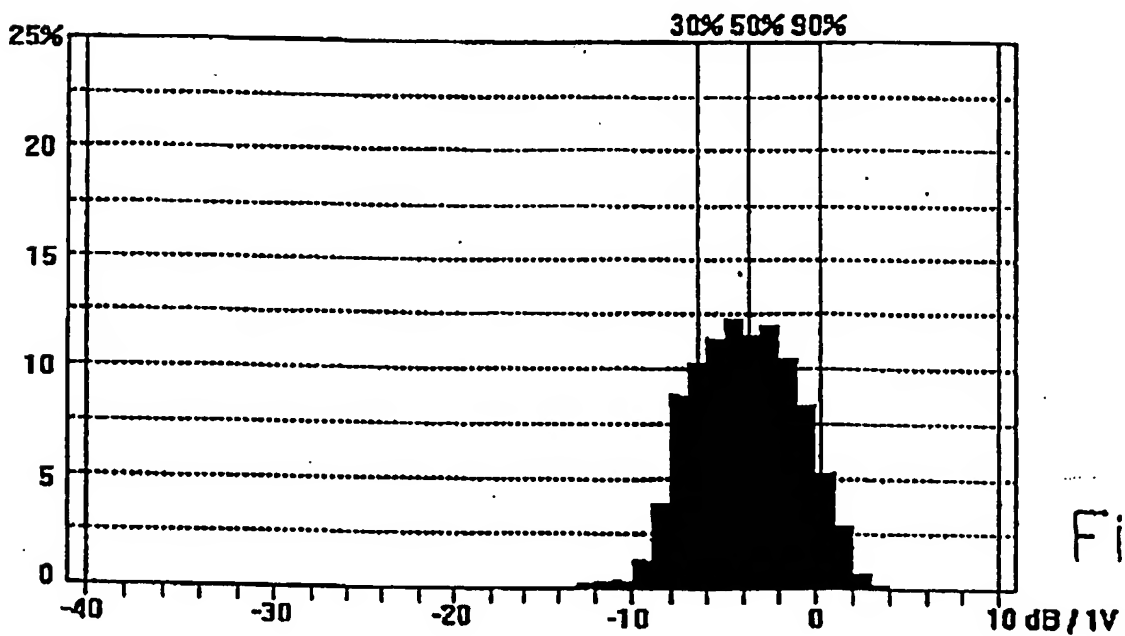


Fig.10

# INTERNATIONAL SEARCH REPORT

International Application No

PCT/DK 99/00531

## A. CLASSIFICATION OF SUBJECT MATTER

IPC 7 H04R25/00

According to International Patent Classification (IPC) or to both national classification and IPC

## B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 7 H04R G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

WPI Data, PAJ

## C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 4 628 529 A (BORTH DAVID E ET AL) 9 December 1986 (1986-12-09) column 1, line 6-10 column 2, line 38-42 column 2, line 50 -column 3, line 24 column 7, line 9 -column 9, line 36 ---	1-4
Y	WO 98 27787 A (TOEPHOLM & WESTERMANN ;BAEKGAARD LARS (DK)) 25 June 1998 (1998-06-25) page 4, line 17 -page 5, line 21 ---	1-4
A	US 4 852 175 A (KATES JAMES M) 25 July 1989 (1989-07-25) cited in the application column 2, line 30 -column 6, line 21 -----	1-4

☐ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

### \* Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

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Date of the actual completion of the international search

30 June 2000

Date of mailing of the international search report

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# INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No

PCT/DK 99/00531

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